Claims:

1. A method of auto-calibrating a surround sound system, comprising the acts of:

producing an electric calibration signal, said calibration signal being a temporal maximum length sequence (MLS) signal,

supplying said calibration signal to an electro-acoustic converter for converting the calibration signal to an acoustic response,

transmitting the acoustic response as a sound wave in a listening environment to an acousto-electric converter for converting the acoustic response received by the acousto-electric converter to an electric response signal,

correlating the electric response signal with the electric calibration signal to compute filter coefficients, and

processing the filter coefficients together with a predetermined channel response of the electro-acoustic converter to produce a substantially whitened system response.

2. The method of claim 1, wherein the acoustic response is radiated in the listening environment for a time less than approximately 3 seconds.

3. The method of claim 1, wherein the surround sound system includes a plurality of audio channels, with each channel having at least one electro-acoustic converter, wherein the substantially whitened response is produced

independently for each audio channel.

4. A method of producing a matched filter for whitening an audio channel in a listening environment, comprising:

producing in the audio channel a test output sound corresponding to a temporal maximum length sequence (MLS) signal,

receiving the test output sound at a predetermined location in the listening environment, thereby producing an impulse response,

analyzing a correlation between the impulse response and the MLS signal, and

generating from the analyzed correlation filter coefficients of the matched filter.

5. The method of claim 4, wherein analyzing the correlation includes producing a polynomial model of the impulse response.

6. The method of claim 4, wherein analyzing the correlation includes using an auto regressive (AR) model.

7. The method of claim 5, wherein generating the filter coefficients includes optimizing a closeness of fit between the polynomial model and the matched filter.

8. The method of claim 7, wherein optimizing the closeness of fit includes adjusting a length of the MLS signal.

9. The method of claim 5, further comprising cascading the matched filter with a useful audio signal so as to produce the substantially whitened audio channel.

10. An auto-calibrating surround sound (ACSS) system, comprising:

an electro-acoustic converter disposed in an audio channel and adapted to emit a sound signal in response to an electric input signal,

a processor generating a test signal represented by a temporal maximum length sequence (MLS) and supplying the test signal as the electric input signal to the electro-acoustic converter, and

an acousto-electric converter receiving the sound signal in a listening environment and supplying a received electric signal to the processor,

wherein the processor correlates the received electric signal with the test signal and determines from the correlated signals a substantially whitened response of the audio channel in the listening environment.

11. The ACSS system of claim 10, wherein the processor includes an impulse modeler that produces a polynomial least-mean-square (LMS) error fit between a desired whitened response and the substantially whitened response determined from the correlated signals.

12. The ACSS system of claim 10, further comprising a coefficient extractor which generates filter coefficients of a corrective filter to produce the substantially whitened response of the audio channel.

13. The ACSS system of claim 12, wherein the corrective filter is located in an audio signal path between an audio signal line input and the electro-acoustic converter and cascaded with the audio signal line input.

14. The ACSS system of claim 12, wherein at least one of the correlator, the IM, and the corrective filter form a part of the processor.

15. The ACSS system of claim 13, wherein the processor is a digital signal processor (DSP).

16. The ACSS system of claim 15, further including an analog-to-digital (A/D) converter that converts an analog audio line input and the electric signal supplied by the acousto-electric converter into temporal digital signals.

17. The ACSS system of claim 15, further including a digital-to-analog (D/A) converter that converts digital output signals from the DSP to an analog audio line output for driving the electro-acoustic converter.

18. A digital filter for whitening an audio channel in a listening environment, comprising:

an input receiving a digital audio signal,

a corrective filter having filter coefficients determined in the listening environment using a maximum length sequence (MLS) test signal, the corrective filter convolving the filter coefficients with the digital audio signal to form a corrected audio signal, and

an output supplying the corrected audio signal to a sound generator.